JOHN ATKINSON

Chord Electronics Hugo M Scaler

UPSAMPLING DIGITAL PROCESSOR

The idea of using digital signal processing (DSP) to convert digital audio data sampled at 44.1kHz or 48kHz to a higher sample rate is not new. I first heard the beneficial effects of upsampling at *Stereophile*'s 1998 hi-fi show in Los Angeles, where a pro-audio dCS 972 digital-to-digital processor[1] was being used to convert 16-bit/44.1kHz CD data to a 24/192 datastream. So persuaded was I of the sonic improvement offered by upsampling that I bought a dCS 972 to upsample CDs to 24/88.2 to feed my then-reference Mark Levinson No.30.6 Reference DAC. As D/A processors improved, my need for upsampling faded away. (While my dCS 972 still sees some use, this is for the opposite purpose: downsampling my hi-rez recordings to 16/44.1 to produce CD masters.)

The M Scaler makes it possible for the accompanying DAC to more accurately reconstruct the analog signal.

I subsequently wrote that I was convinced that the sonic improvement I heard with the dCS 972 was due to its using a different oversampling digital reconstruction filter with a different number of taps and, as a consequence, different passband ripple and stopband rejection.[2] Twenty years after those words appeared in print, the Chord Electronics $4795 Hugo M Scaler arrived in my listening room, and I found myself returning to the subject of digital filters and upsampling—or upscaling, as the British company calls it.

Description

The Hugo M Scaler is a small, almost square component powered from a supplied 15V wall wart and controlled with either a small remote or the front-panel buttons. It features five digital inputs: a galvanically isolated Type-B USB, two coaxial S/PDIF on BNC, and two TosLink optical. (DSD data is converted to PCM with a 6dB reduction in level.) The M Scaler doesn't have analog outputs, but it has three digital outputs: one coaxial BNC S/PDIF, one optical, and a pair of galvanically isolated BNC jacks that enable upsampling to 705.6kHz or 768kHz—but only when used with compatible Chord Electronics DACs. The M Scaler will work with D/A processors from other manufacturers, but only, of course, up to the maximum frequency the DAC can accept. (My PS Audio DirectStream DAC indicated it was receiving data sampled at 384kHz when I set the M Scaler to output that rate via an S/PDIF connection, but the DirectStream is limited to 192kHs and there was no sound.)

The front panel offers six of Chord's traditional glass-sphere buttons, which illuminate in different colors accord-

1 Jonathan Scull reviewed the dCS 972 in February 1999; see stereophile.com/digitalprocessors/260/index.html. dCS subsequently introduced a functionally identical but cosmetically improved consumer version, the Purcell; see stereophile.com/digitalprocessors/454/index.html.

2 See stereophile.com/asweseeit/344/index.html. In a letter responding to this essay, Bob Katz conjectured that because the inevitable quantization distortion is spread over a wider frequency range with upsampled data, the audibility of this distortion is significantly reduced.

SPECIFICATIONS

Description Upsampling digital processor with adjustable output sample rate. Digital inputs: 2 coaxial (BNC), 2 optical (TosLink), 1 USB (Type B). Digital outputs: dual-BNC, BNC, TosLink optical. PCM formats supported: 44.1–768kHz, 16/24/32-bit. Included accessories: 15V/4A switching power supply, IR remote control, dual-BNC cables.

Dimensions 9.25" (235mm) W × 1.6" (40.5mm) H × 9.4" (238mm) D. Weight: 5.6lb (2.55kg).

Finishes Silver, Black.

Serial number of unit reviewed 42051, "Made In England."

Price $4795. Approximate number of dealers: 100, 35 of whom stock the M Scaler. Warranty: three years.

Manufacturer Chord Electronics Ltd., The Pumphouse, Farleigh Bridge, Farleigh Lane, East Farleigh, Kent ME16 9NB, England, UK. Tel: (44) (0)1622-721444. Web: chordelectronics.co.uk. US distributor: Bluebird Music Ltd., 275 Woodward Avenue, Buffalo, NY 14217. Tel: (416) 638-8207. Web: bluebirdmusic.com.

ing to what the M Scaler has been asked to do. Only four of these are currently functional; the rightmost pair, marked "DX," are intended for a future product design. From left to right, these indicate: whether video mode or automatic input detection is selected; which input is in use; the output sample rate; and the input sample rate.

The M Scaler offers a video mode, as the upsampling introduces a latency up to 600ms, too long for synchronizing audio with video. In video mode, the M Scaler uses an interpolation filter with lower latency. For audio use, a passthrough option, with the output sample rate the same as the input rate, is provided to allow comparison to upsampled output. However, as the upsampling can introduce digital "overs"—interpolated sample values exceeding 0dBFS—both the passthrough and upsampled signals are reduced in gain by about 2.8dB, which complicates such direct comparisons.

The core of the M Scaler is a Xilinx XC7A200T FPGA (field-programmable gate array) on which runs the code for the Watts Transient Alignment reconstruction filter—named for design consultant Rob Watts—first seen in Chord's Blu Mk.II upscaling CD transport. The FPGA has 740 DSP cores; Watts's filter uses 528 of them running in parallel at 16Fs and a bit depth of 56 to achieve a filter length of an extraordinary 1,015,808 taps. For comparison, the WTA filter in Chord's DAVE D/A processor, which I reviewed in June 2017,[3] used 164,000 taps implemented in 166 DSP cores.

In preparing the DAVE review, I asked Watts what is the advantage of using ever-longer digital filters. "If you have a conventional filter with 100 taps, you will recover *some* of the transient information," he explained. "A 100-tap filter gives you sufficiently good *frequency*-domain performance, but not in the time domain. . . . Every time you increase the number of taps, you improve the perception of pitch, timbre gets better—bright instruments sound brighter, dark instruments sound darker—the starting and stopping of notes becomes easier to hear, the localization of sounds get better. There is less listening fatigue—the brain has to do less processing of the information presented to it to understand what's going on."

Digital filters and upsampling

In the promotional literature for the M Scaler,[4] Chord writes, "The Hugo M Scaler . . . takes the digital file and repairs it, adding back the information lost between the samples, then it sends the repaired file to the DAC. . . . With 705,600 samples per second, a huge amount of important information that was lost when creating the 44.1 digital file is now recovered. The more samples, the closer you get

3 See stereophile.com/content/chord-electronics-dave-da-processor.

4 See chordelectronics.co.uk/product/hugo-mscaler.

MEASUREMENTS

It was difficult to decide which tests would be best at uncovering what the Chord Hugo M Scaler was doing. If I examined, with my Audio Precision SYS2722 system (see the January 2008 "As We See It"[1]), the analog output of Chord's DAVE or another D/A processor when fed upsampled data from the M Scaler, the result would be obscured by the anti-aliasing filter of the AP's A/D converter. I could examine the digital output of the M Scaler, but the analyzer's S/PDIF input is restricted to sample rates of 192kHz and below. Despite these cautions, I feel the following measurements do characterize the M Scaler's behavior.

For the testing, I fed data sampled at 44.1kHz or 48kHz to one of the M Scaler's optical inputs from either the Audio Precision's optical S/PDIF output or from that of a Pioneer SACD player used as a transport. I also fed the Chord USB data sourced from my MacBook Pro running on battery power with Pure Music 3.0 playing WAV and AIFF test-tone files. Apple's USB Prober utility identified the Chord as "Hugo M SCALER" from "Chord Electronics Ltd" with the serial number "413-001." The M Scaler's USB port operated in the optimal isochronous asynchronous mode. Apple's AudioMIDI utility revealed that, via USB, the M Scaler accepted 32-bit integer data sampled at all rates from 32 to 768kHz.

In the M Scaler's pass-through mode, setting the output sample rate to be the same as the input rate reduces the signal level and adds dither at the LSB level. The blue trace in fig.1 shows the Scaler's impulse response, captured in the digital

1 See http://tinyurl.com/4ffpve4.

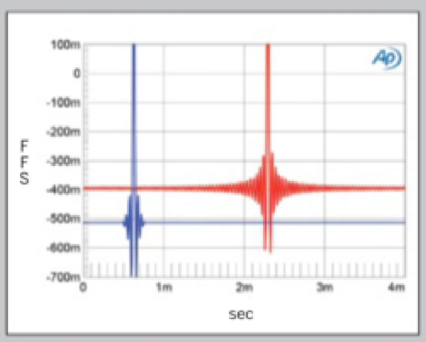

Fig.1 Chord M Scaler, digital-domain impulse response sampled at 48kHz, resampled to 48kHz (blue) and upsampled to 192kHz (red) (4ms time window, exaggerated vertical scale).

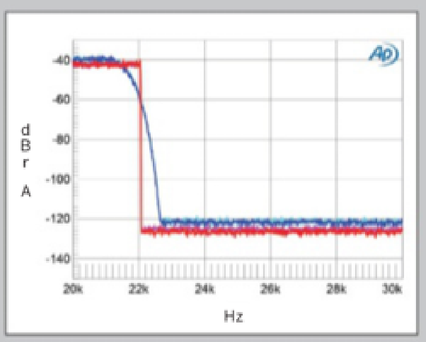

Fig.2 Mark Levinson No.30.6, spectrum, 20kHz–30kHz, of 16-bit white noise sampled at 44.1kHz at –4dBFS (left channel blue, right cyan) and upsampled to 88.2kHz with M Scaler (left channel red, right magenta) (20dB/vertical div.).

to the original analog signal. . . . The Hugo M Scaler in essence places 15 additional new musical samples in between each original musical sample, resulting in an astounding improvement in the recreation of the original music signal."

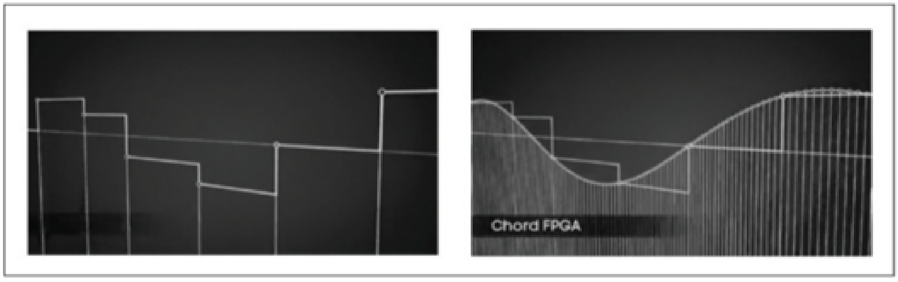

Fig.1 From the Chord Hugo M Scaler white paper: CD sample-rate sample values (left); sample values after 16× upsampling (right).

My eyebrows raised, I kept reading. Referring to the figure reprinted here, the text states that "The Hugo M Scaler takes a rough stairstep CD quality waveform and transforms it into a smooth analog-like waveform. That quantum leap in sampling brings a breathtaking leap in detail, accuracy and realism to your music."

Hmm. The measurements I performed to accompany our reviews of the dCS 972 and Purcell definitively showed that upsampling *doesn't* add information above the Nyquist frequency—22.05kHz with CD data—of the lower sample rate. So what is the M Scaler doing?

In one of the first articles I wrote for *Stereophile*, "Zen & The Art of D/A Conversion," which was published in September 1986,[5] I discussed how the recovered analog signal is not directly described by the levels of the digital samples. Instead, the *interaction* between those samples and the impulse response of a digital low-pass reconstruction filter recreates the analog waveform—not just at the sampling intervals but *between* them.[6] By processing the incoming data with a low-pass filter featuring an extremely long impulse response, the M Scaler makes it possible for the accompanying DAC to more accurately reconstruct the analog signal. In effect, it replaces the DAC's digital filter with its own, as the DAC's filter is now operating at the higher sample rate, and its cutoff is one or more octaves above the original data's Nyquist frequency.

Listening with the Chord Electronics DAVE

I performed most of my auditioning of the M Scaler with a sample of Chord's DAVE D/A processor, sourcing audio data from my Roon Nucleus+ server via USB and sending the upsampled data to the DAVE with a dual-BNC connection. I also used the M Scaler with my PS Audio DirectStream and Mark Levinson No.30.6 DACs, using a single S/PDIF connection. I tried using the M Scaler with my NAD M10 integrated amplifier, sending the latter data upsampled to 96kHz or 192kHz via an optical S/PDIF link. However, while the NAD would play music for a few seconds, it stuttered and then stopped. I was using the M10's Dirac room correction and wondered if that was the problem, so switched off the Dirac filter, but there was no improvement.

In my DAVE review, I concluded that that DAC's "superb re-creation of soundstage depth, its sense of musical drive, and the clarity with which it presented recorded detail were addictive." Listening to this new sample of the DAVE, that is what

5 See stereophile.com/reference/25/index.html.

6 For a detailed explanation of how a digital filter operates, see stereophile.com/content/zen-art-ad-conversion.

measurements, continued

domain (no conversion to analog) when fed a 48kHz impulse and set to output 48kHz data. The vertical scale in this graph is exaggerated to reveal fine detail—the impulse is typical of a short linear-phase filter, with just three samples of pre- and post-ringing. By contrast, the red trace in fig.1 shows the effect of upsampling the 48kHz impulse to 192kHz, again captured in the digital domain. The filter is still a time-symmetrical linear-phase type, but now a very large amount of pre- and post-ringing is present.

To look at the effect of the M Scaler's upsampling filter in the frequency domain, I wanted to use a conventional D/A processor. I dug out the sample of the Mark Levinson No.30.6 that I purchased after reviewing it back in 1999. I first fed it 16-bit white noise sampled at 44.1kHz. The blue and cyan traces in fig.2 show the effect of the Levinson's reconstruction filter with this signal, plotted between 20kHz and 30kHz and with the processor's analog output signal captured by Audio Precision's A/D converter running at a 100kHz sample rate. The output starts to roll off above 21kHz and reaches full stop-band attenuation at 22.7kHz.

The red trace shows the spectrum of the No.30.6's analog output when fed the same data upsampled to 88.2kHz by the M Scaler. The output is now flat to 22kHz, and the Chord's upsampling filter drops like a stone above that frequency, reaching full attenuation at exactly half the original sampling frequency, 22.05kHz. The 2.8dB reduction in level imposed by the M Scaler DSP can be seen below 21kHz in this graph. However, for reasons I can't figure, the reduction in the upsampled ultrasonic stop-band noise is twice as large.

I repeated these spectral analyses with other signals, including single- and multitone signals. The only sign of mild misbehavior was with a full-scale, 24-bit 20kHz tone, sampled at 48kHz and upsampled to 192kHz with the M Scaler (fig.3). Analyzed in the digital domain—no D/A conversion—aliased images can be seen at 28kHz (48–20), 68kHz (48+20), and 76kHz (96–20). However, as these all lie at or below –140dBFS (0.00001%), effectively they won't exist in real-world D/A conversion!—**John Atkinson**

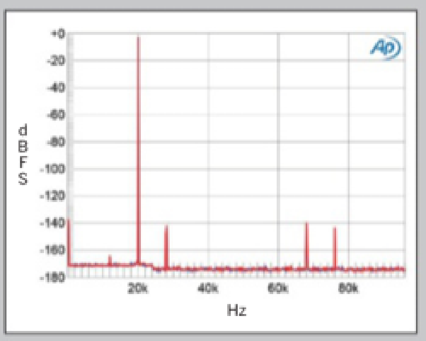

Fig.3 Chord M Scaler, wideband digital-domain spectrum, 24-bit 20kHz tone at 0dBFS sampled at 48kHz and upsampled to 192kHz (20dB/vertical div.).

I heard. But with the DAVE working with CD data upsampled to the maximum rate of 705.6kHz or 768kHz, there was now even more image depth, an increased sense of drive, and even more clarity. These improvements were not just audible with the magnificent Magico M2 full-range speakers; I also heard them with the KEF LS50 minimonitors.

Back in 2017, I had used the DAVE to audition various options for dither and noise-shaping when I prepared the 16/44.1 master for *Stereophile's Tight Lines* CD. As coincidence would have it, midway through the time I had the DAVE and M Scaler in my system, I was preparing the CD master for *Translation*, an album of works by modern Latvian composer Ēriks Ešenvalds, performed by the Portland State Chamber Choir directed by Ethan Sperry and produced by Erick Lichte. We had recorded the sessions at 24/96, so I was downsampling the hi-rez data with my dCS 972, and again I was using the DAVE to audition the various options. Once I decided on the optimal strategy, but before I sent the master to Naxos, I tried upsampling the CD files with the M Scaler to see how they compared with the hi-rez originals.

Ešenvalds composes, not just with what the singers sing and the musicians play, but also with the spatial relationships between the performers. In the final work on the album, *In Paradisum*, the composer contrasts a downstage solo cello with a distant, almost off-stage viola, set within a mostly wordless, vocalise halo from the choir. With the original 24/96 file, the spatial settings of the viola, cello, and choir were unambiguous in both width and depth. When I upsampled the CD version to 24/705.6, I was hard put to hear any difference between it and the 24/96 original! Perhaps there was a little less soundstage depth? It was only when I upscaled the 24/96 data to 24/768 that I felt it moved ahead of the upsampled CD version, with even better spatial differentiation of the acoustic objects.

ASSOCIATED EQUIPMENT

Digital sources Roon Nucleus+ file server; Ayre Acoustics C-5xe[MP] universal player; Chord DAVE, PS Audio PerfectWave DirectStream, Mark Levinson No.30.6 D/A processors.
Preamplifier NHT PVC balanced volume control (with Mark Levinson).
Power amplifiers Lamm Industries M1.2 Reference monoblocks.
Integrated amplifier NAD M10.
Loudspeakers Magico M2, KEF LS50, Vimberg Mino.
Cables Digital: AudioQuest Vodka (Ethernet), Belkin Gold (USB), Chord (dual-BNC connection to DAVE), Esperanto Blue (S/PDIF to PS Audio), AudioQuest OptiLink (optical S/PDIF to Mark Levinson and NAD). Interconnect: AudioQuest Wild Blue (balanced). Speaker: AudioQuest K2. AC: AudioQuest Dragon Source & High Current, manufacturers' own.
Accessories Target TT-5 equipment racks; Ayre Acoustics Myrtle Blocks; Celestion 24" stands (with KEFs); ASC Tube Traps, RPG Abffusor panels; AudioQuest Niagara 1000 (source components) and 5000 (amplifiers) Low-Z Power/Noise-Dissipation Systems. AC power comes from two dedicated 20A circuits, each just 6' from breaker box.—**John Atkinson**

Listening with third-party processors

Last December, I was held spellbound by Russian pianist Anna Gourari performing solo the *Adagio* from J.S. Bach's Concerto in D Minor, BWV 974. When I got home from the concert, I called up Ms. Gourari's *Elusive Affinity* album on Tidal (24/48 MQA FLAC, ECM 2612) and streamed it directly to the PS Audio PerfectWave DirectStream DAC. While the recording equals the live concert version in musicality, with Roon unfolding the MQA-encoded data to 96kHz, the piano's right-hand register was a little too far forward in the mid-treble. Upsampling the Roon-unfolded data to 192kHz with the M Scaler reduced the slight clanginess to the sound of the piano and better presented the relationship between the piano and the subtle acoustic of the record venue.

I heard the same improvement in the spatial relationships between the instruments and the recording space on "I Heard It Through the Grapevine" from Bill Frisell's *East/West* (16/44.1 ALAC file, ripped from Nonesuch 7559798632), when I compared the 24/176.4–upsampled data with the original data sent directly to the PS Audio. This was particularly noticeable with Kenny Wollesen's drums, which moved back in the image when I upscaled the data with the Chord, cleaning the window into the recorded soundstage.

My final listening session was with the M Scaler sending CD data upsampled to 88.2kHz and 24 bits to the Mark Levinson No.30.6.[7] (The two-decades–old Levinson can't handle data with a sample rate greater than 96kHz.) Unlike the PS Audio and Chord DACs, the Levinson doesn't have a volume control, so I inserted an NHT PVC balanced passive control immediately in front of the Lamm's inputs.

It's been a long time since I last listened to the No.30.6. Feeding it the Frisell "Grapevine" with the M Scaler in pass-through mode, I was struck by the low-frequency authority and control it exerted on the double bass and kickdrum, as well as the sense of musical momentum. But the highs sounded grainier than they had with the PS Audio or DAVE, and there was less image depth. Sending the No.30.6 upsampled 24/88.2 data, the bass remained authoritative; the drums moved farther back in the soundstage; Frisell's electric guitar moved in front of the plane of the speakers; the sounds of the various stereo effects he uses moved both forward and beyond the speaker positions; and the treble lost much of its grain. Nice. Very nice. The Hugo M Scaler brought the performance of the Levinson No.30.6 DAC into the third decade of this century.

Conclusions

As David Rich, then with *The Audio Critic*, wrote in the 1990s, "in the next century, all audiophiles will be listening to will be different digital filters." Chord's Hugo M Scaler illustrates Dr. Rich's point: It replaces the various reconstruction filters used in other manufacturers' DACs with Rob Watts's enormously long WTA filter. That filter does sound superb, and, as a bonus—in addition to upgrading the sounds of older DACs—the M Scaler adds a USB input with Roon compatibility to DACs that don't have one, like my Levinson. At $4795, the M Scaler is relatively expensive; I recommend you audition the M Scaler with your own DAC before getting out the credit card. But "[improve] the recreation of the original music signal," as Chord claims, the M Scaler definitely did, with all three D/A processors I tried. ■

7 See stereophile.com/digitalprocessors/159/index.html.